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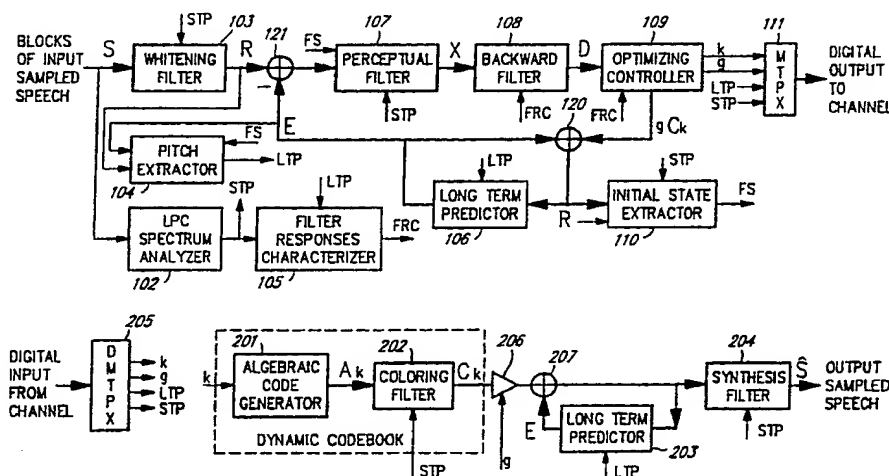
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(54) Title: DYNAMIC CODEBOOK FOR EFFICIENT SPEECH CODING BASED ON ALGEBRAIC CODES



(57) Abstract

A method of encoding a speech signal is disclosed. This method improves the excitation codebook and search procedure of the conventional Code Excited Linear Prediction (CELP) speech encoders. Use is made of a dynamic codebook (201, 202) based on a combination of two modules: a sparse algebraic code generator (201) associated to a filter (202) having a transfer function varying in time. The generator (201) is a structured codebook with codewords having very few non zero components. The filter (202) shapes the spectral characteristics whereby the resulting excitation codebook (201, 202) exhibits favorable perceptual properties. The search complexity in finding the best codeword is greatly reduced by bringing the search back to the algebraic code domain thereby allowing the sparsity of the algebraic code to speed up the necessary computations.

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DYNAMIC CODEBOOK FOR EFFICIENT SPEECHCODING BASED ON ALGEBRAIC CODES

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BACKGROUND OF THE INVENTION1. Field of the invention:

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The present invention relates to a new technique for digitally encoding and decoding in particular but not exclusively speech signals in view of transmitting and synthesizing these speech signals.

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2. Brief description of the prior art:

Efficient digital speech encoding techniques with good subjective quality/bit rate tradeoffs are increasingly in demand for numerous applications such as voice transmission over satellites, land mobile, digital radio or packed network, for voice storage, voice response and secure telephony.

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One of the best prior art methods capable of achieving a good quality/bit rate tradeoff is the so called Code Excited Linear Prediction (CELP) technique. In accordance with this method, the speech signal is sampled and converted into successive blocks

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of a predetermined number of samples. Each block of samples is synthesized by filtering an appropriate innovation sequence from a codebook, scaled by a gain factor, through two filters having transfer functions varying in time. The first filter is a Long Term Predictor filter (LTP) modeling the pseudoperiodicity of speech, in particular due to pitch, while the second one is a Short Term Predictor filter (STP) modeling the spectral characteristics of the speech signal. The encoding procedure used to determine the parameters necessary to perform this synthesis is an analysis by synthesis technique. At the encoder end, the synthetic output is computed for all candidate innovation sequences from the codebook. The retained codeword is the one corresponding to the synthetic output which is closer to the original speech signal according to a perceptually weighted distortion measure.

The first proposed structured codebooks are called stochastic codebooks. They consist of an actual set of stored sequences of N random samples. More efficient stochastic codebooks propose derivation of a codeword by removing one or more elements from the beginning of the previous codeword and adding one or more new elements at the end thereof. More recently, stochastic codebooks based on linear combinations of a small set of stored basis vectors have greatly reduced the search complexity. Finally, some algebraic structures have also been proposed as excitation codebooks with efficient search procedures.

However, the latter are designed for speed and they lack flexibility in constructing codebooks with good subjective quality characteristics.

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OBJECTS OF THE INVENTION

10 The main object of the present invention is to
 combine an algebraic codebook and a filter with a
 transfer function varying in time, to produce a
 dynamic codebook offering both the speed and memory
 saving advantages of the above discussed structured
15 codebooks while reducing the computation complexity of
 the Code Excited Linear Prediction (CELP) technique
 and enhancing the subjective quality of speech.

SUMMARY OF THE INVENTION

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 More specifically, in accordance with the
 present invention, there is provided a method of
 producing an excitation signal that can be used in
25 synthesizing a sound signal, comprising the steps of
 generating a codeword signal in response to an index
 signal associated to this codeword signal, such signal
 generating step using an algebraic code to generate
 the codeword signal, and filtering the so generated
30 codeword signal to produce the excitation signal.

Advantageously, the algebraic code is a sparse algebraic code.

5 The subject invention also relates to a dynamic codebook for producing an excitation signal that can be used in synthesizing a sound signal, comprising means for generating a codeword signal in response to an index signal associated to this codeword signal, which signal generating means using an algebraic code
10 to generate the codeword signal, and means for filtering the so generated codeword signal to produce the excitation signal.

15 In accordance with a preferred embodiment of the dynamic codebook, the filtering means comprises a coloring filter having a transfer function varying in time to shape the frequency characteristics of the excitation signal so as to damp frequencies perceptually annoying the human ear. This coloring
20 filter comprises an input supplied with linear predictive coding parameters representative of spectral characteristics of the the sound signal to vary the above mentioned transfer function.

25 In accordance with other aspects of the present invention, there is also provided:

30 (1) a method of selecting one particular algebraic codeword that can be processed to produce a signal excitation for a synthesis means capable of synthesizing a sound signal, comprising the steps of

(a) whitening the sound signal to be synthesized to generate a residual signal, (b) computing a target signal X by processing a difference between the residual signal and a long term prediction component of the signal excitation, (c) backward filtering the target signal to calculate a value D of this target signal in the domain of an algebraic code, (d) calculating, for each codeword among a plurality of available algebraic codewords A_k expressed in the algebraic code, a target ratio which is function of the value D , the codeword A_k , and a transfer function $H = D / X$, and (e) selecting the said one particular codeword among the plurality of available algebraic codewords in function of the calculated target ratios.

(2) an encoder for selecting one particular algebraic codeword that can be processed to produce a signal excitation for a synthesis means capable of synthesizing a sound signal, comprising (a) means for whitening the sound signal to be synthesized and thereby generating a residual signal, (b) means for computing a target signal X by processing a difference between the residual signal and a long term prediction component of the signal excitation, (c) means for backward filtering the target signal to calculate a value D of this target signal in the domain of an algebraic code, (d) means for calculating, for each codeword among a plurality of available algebraic codewords A_k expressed in the above mentioned algebraic code, a target ratio which is function of the value D , the codeword A_k , and a transfer function

H = D / X , and (e) means for selecting the said one particular codeword among the plurality of available algebraic codewords in function of the calculated target ratios. In accordance with preferred
5 embodiments of the encoder, the target ratio comprises a numerator given by the expression $P^2(k) = (DAk^T)^2$ and a denominator given by the expression $\alpha^2 k = \|AkH^T\|^2$, where Ak and H are under the form of matrix, each codeword Ak is a waveform comprising a small number of
10 non-zero impulses each of which can occupy different positions in the waveform to thereby enable composition of different codewords, the target ratio calculating means comprises means for calculating into a plurality of embedded loops contributions of the
15 non-zero impulses of the considered algebraic codeword to the numerator and denominator and for adding the so calculated contributions to previously calculated sum values of these numerator and denominator, respectively, the embedded loops comprise an inner
20 loop, and the codeword selecting means comprises means for processing in the inner loop the calculated target ratios to determine an optimized target ratio and means for selecting the said one particular algebraic codeword in function of this optimized target ratio.

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(3) a method of generating at least one long term prediction parameter related to a sound signal in view of encoding this sound signal, comprising the steps of (a) whitening the sound signal to generate a
30 residual signal, (b) producing a long term prediction component of a signal excitation for a synthesis means

component of a signal excitation for a synthesis means capable of synthesizing the sound signal, which producing step including estimating an unknown portion of the long term prediction component with the residual signal, and (c) calculating the long term prediction parameter in function of the so produced long term prediction component of the signal excitation.

(4) a device for generating at least one long term prediction parameter related to a sound signal in view of encoding this sound signal, comprising (a) means for whitening the sound signal and thereby generating a residual signal, (b) means for producing a long term prediction component of a signal excitation for a synthesis means capable of synthesizing the sound signal, these producing means including means for estimating an unknown portion of the long term prediction component with the residual signal, and (c) means for calculating the long term prediction parameter in function of the so produced long term prediction component of the signal excitation.

The objects, advantages and other features of the present invention will become more apparent upon reading of the following, non restrictive description of a preferred embodiment thereof, given with reference to the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

In the appended drawings:

5

Figure 1 is a schematic block diagram of the preferred embodiment of an encoding device in accordance with the present invention;

10

Figure 2 is a schematic block diagram of a decoding device using a dynamic codebook in accordance with the present invention;

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Figure 3 is a flow chart showing the sequence of operations performed by the encoding device of Figure 1;

20

Figure 4 is a flow chart showing the different operations carried out by a pitch extractor of the encoding device of Figure 1, for extracting pitch parameters including a delay T and a pitch gain b ; and

25

Figure 5 is a schematic representation of a plurality of embedded loops used in the computation of optimum codewords and code gains by an optimizing controller of the encoding device of Figure 1.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

5 Figure 1 is the general block diagram of a
speech encoding device in accordance with the present
invention. Before being encoded by the device of
Figure 1, an analog input speech signal is filtered,
typically in the band 200 to 3400 Hz and then sampled
10 at the Nyquist rate (e.g. 8 kHz). The resulting
signal comprises a train of samples of varying
amplitudes represented by 12 to 16 bits of a digital
code. The train of samples is divided into blocks
which are each L samples long. In the preferred
15 embodiment of the present invention, L is equal to 60.
Each block has therefore a duration of 7.5 ms. The
sampled speech signal is encoded on a block by block
basis by the encoding device of Figure 1 which is
broken down into 10 modules numbered from 102 to 111.
20 The sequence of operation performed by these modules
will be described in detail hereinafter with reference
to the flow chart of Figure 3 which presents numbered
steps. For easy reference, a step number in Figure 3
and the number of the corresponding module in Figure
25 1 have the same last two digits. Bold letters refer
to L-sample-long blocks (i.e. L-component vectors).
For instance, **S** stands for the block [S(1),
S(2), ..., S(L)].

30 Step 301: The next block **S** of L samples is supplied to
the encoding device of Figure 1.

Step 302: For each block of L samples of speech signal, a set of Linear Predictive Coding (LPC) parameters, called STP parameters, is produced in accordance with a prior art technique through an LPC spectrum analyser 102. More specifically, the latter analyser 102 models the spectral characteristics of each block s of samples. In the preferred embodiment, the parameters STP comprise a number M=10 of prediction coefficients [a1, a2,...aM]. One can refer to the book by J.D. Markel & A.H. Gray, Jr: "Linear Prediction of Speech" Springer Verlag (1976) to obtain information on representative methods of generating these parameters.

Step 303: The input block s is whitened by a whitening filter 103 having the following transfer function based on the current values of the STP prediction parameters:

$$A(z) = \sum_{i=0}^M a_i z^{-i} \quad (1)$$

where $a_0 = 1$, and z represents the variable of the polynomial A(z).

As illustrated in Figure 1, the filter 103 produces a residual signal R.

Of course, as the processing is performed on a block basis, unless otherwise stated, all the filters are assumed to store their final state for use as initial state in the following block processing.

5

The purpose of step 304 is to compute the speech periodicity characterized by the Long Term Prediction (LTP) parameters including a delay T and a pitch gain b .

10

Before further describing step 304, it is useful to explain the structure of the speech decoding device of Figure 2 and understand the principle upon which speech is synthesized.

15

As shown in Figure 2, a demultiplexer 205 interprets the binary information received from a digital input channel into four types of parameters, namely the parameters STP, LTP, k and g . The current block s of speech signal is synthesized on the basis of these four parameters as will be seen hereinafter.

20

The decoding device of Figure 2 follows the classical structure of the CELP (Code Excited Linear Prediction) technique insofar as modules 201 and 202 are considered as a single entity: the (dynamic) codebook. The codebook is a virtual (i.e. not actually stored) collection of L -sample-long waveforms (codeword) indexed by an integer k . The index k ranges from 0 to $NC-1$ where NC is the size of the codebook. This size is 4096 in the preferred

25

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embodiment. In the CELP technique, the output speech signal is obtained by first scaling the k^{th} entry of the codebook by the pitch gain g through an amplifier 206. An adder 207 adds the so obtained scaled waveform, gC_k , to the output E (the long term prediction component of the signal excitation of a synthesis filter 204) of a long term predictor 203 placed in a feedback loop and having a transfer function $B(z)$ defined as follows:

10

$$B(z) = bz^{-T} \quad (2)$$

15 where b and T are the above defined pitch gain and delay, respectively.

The predictor 203 is a filter having a transfer function influenced by the last received LTP parameters b and T to model the pitch periodicity of speech. It introduces the appropriate pitch gain b and delay of T samples. The composite signal $gC_k + E$ constitutes the signal excitation of the synthesis filter 204 which has a transfer function $1/A(z)$. The filter 204 provides the correct spectrum shaping in accordance with the last received STP parameters. More specifically, the filter 204 models the resonant frequencies (formants) of speech. The output block 8 is the synthesized (sampled) speech signal which can be converted into an analog signal with proper anti-

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aliasing filtering in accordance with a technique well known in the art.

In the present invention, the codebook is
5 dynamic; it is not stored but is generated by the two
modules 201 and 202. In a first step, an algebraic
code generator 201 produces in response to the index
k and in accordance with a Sparse Algebraic Code (SAC)
a codeword A_k formed of a L-sample-long waveform
10 having very few non zero components. In fact, the
generator 201 constitutes an inner, structured
codebook of size N_C . In a second step, the codeword
 A_k from the generator 201 is processed by a coloring
filter 202 whose transfer function $F(z)$ varies in time
15 in accordance with the STP parameters. The filter 202
colors, i.e. shapes the frequency characteristics
(dynamically controls the frequency) of the output
excitation signal C_k so as to damp a priori those
frequencies perceptually more annoying to the human
20 ear. The excitation signal C_k , sometimes called the
innovation sequence, takes care of whatever part of
the original speech signal left unaccounted by either
the above defined formant and pitch modelling. In the
preferred embodiment of the present invention, the
25 transfer function $F(z)$ is given by the following
relationship:

$$30 \quad F(z) = \frac{A(z\gamma_1^{-1})}{A(z\gamma_2^{-1})} \quad (3)$$

where $\gamma_1 = .7$ and $\gamma_2 = .85$.

There are many ways to design the generator 201. An advantageous method consists of interleaving four single-pulse permutation codes as follows. The
5 codewords A_k are composed of four non zero pulses with fixed amplitudes, namely $S_1=1$, $S_2=-1$, $S_3=1$, and $S_4=-1$. The positions allowed for S_i are of the form $p_i=2i+8m_i-1$, where $m_i=0, 1, 2, \dots, 7$. It should be noted that for $m_3=7$ (or $m_4=7$) the position p_3 (or p_4) falls beyond
10 $L=60$. In such a case, the impulse is simply discarded. The index k is obtained in a straightforward manner using the following relationship:

15

$$k = 512 m_1 + 64 m_2 + 8 m_3 + m_4 \quad (4)$$

The resulting A_k -codebook is accordingly
20 composed of 4096 waveforms having only 2 to 4 non zero impulses.

Returning to the encoding procedure, it is useful to discuss briefly the criterion used to select
25 the best excitation signal C_k . This signal must be chosen to minimize, in some ways, the difference $\hat{s} - s$ between the synthesized and original speech signals. In original CELP formulation, the excitation signal C_k is based on a Mean Squared Error (MSE)
30 criteria applied to the error $\Delta = \hat{s}' - s'$, where \hat{s}' , respectively s' , is \hat{s} , respectively s , processed by a

perceptual weighting filter of the form $A(z)/A(z\gamma^{-1})$ where $\gamma = 0.8$ is the perceptual constant. In the present invention, the same criterion is used but the computations are performed in accordance with a backward filtering procedure which is now briefly recalled. One can refer to the article by J.P. Adoul, P. Mabillean, M. Delprat, & S. Morissette: "Fast CELP coding based on algebraic codes", Proc. IEEE Int'l conference on acoustics speech and signal processing, pp 1957-1960 (April 1987), for more details on this procedure. Backward filtering brings the search back to the C_k -space. The present invention brings the search further back to the A_k -space. This improvement together with the very efficient search method used by controller 109 (Figure 1) and discussed hereinafter enables a tremendous reduction in computation complexity with regard to the conventional approaches.

It should be noted here that the combined transfer function of the filters 103 and 107 (Figure 1) is precisely the same as that of the above mentioned perceptual weighting filter which transforms S into S' , that is transforms S into the domain where the MSE criterion can be applied.

25

Step 304: To carry out this step, a pitch extractor 104 (Figure 1) is used to compute and quantize the LTP parameters, namely the pitch delay T ranging from T_{min} to T_{max} (20 to 146 samples in the preferred embodiment) and the pitch gain g . Step 304 itself comprises a plurality of steps as illustrated in

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Figure 4. Referring now to Figure 4, a target signal Y is calculated by filtering (step 402) the residual signal R through the perceptual filter 107 with its initial state set (step 401) to the value FS available from an initial state extractor 110. The initial state of the extractor 104 is also set to the value FS as illustrated in Figure 1. The long term prediction component of the signal excitation, $E(n)$, is not known for the current values $n = 1, 2, \dots$. The values $E(n)$ for $n = 1$ to $L-Tmin+1$ are accordingly estimated using the residual signal R available from the filter 103 (step 403). More specifically, $E(n)$ is made equal to $R(n)$ for these values of n . In order to start the search for the best pitch delay T , two variables Max and r are initialized to 0 and $Tmin$ respectively (step 404). With the initial state set to zero (step 405), the long term prediction part of the signal excitation shifted by the value r , $E(n-r)$, is processed by the perceptual filter 107 to obtain the signal z . The crosscorrelation ρ between the signals Y and z is then computed using the expression in block 406 of Figure 4. If the crosscorrelation ρ is greater than the variable Max (step 407), the pitch delay T is updated to r , the variable Max is updated to the value of the crosscorrelation ρ and the pitch energy term α_p equal to $\|z\|^2$ is stored (step 410). If r is smaller than $Tmax$ (step 411), it is incremented by one (step 409) and the search procedure continues. When r reaches $Tmax$, the optimum pitch gain b is computed and quantized using the expression $b=Max/\alpha_p$ (step 412).

Step 305: In step 305, a filter responses characterizer 105 (Figure 1) is supplied with the STP and LTP parameters to compute a filter responses characterization FRC for use in the later steps. The

5 FRC information consists of the following three components where $n = 1, 2, \dots L$. It should also be noted that the component $f(n)$ includes the long term prediction loop.

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• $f(n)$: impulse response of $F(z) \frac{1}{1-bz^{-1}}$ (5a)

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• $h(n)$: response of $\frac{1}{A(z\gamma^{-1})}$ to $f(n)$ (5b)

20

with zero initial state.

25

• $u(i, j)$: autocorrelation of $h(n)$; i.e.:

30

$$u(i, j) = \sum_{k=1}^L h(k-i+1)h(k-j+1) ; \text{ for } 1 \leq i \leq L \quad (5c)$$

35

and $i \leq j \leq L$; $h(n) = 0$ for $n < 1$

The utility of the FRC information will become obvious upon discussion of the forthcoming steps.

Step 306: The long term predictor 106 is supplied with the signal excitation $E + gC_k$ to compute the component E of this excitation contributed by the long term prediction (parameters LTP) using the proper pitch delay T and gain b . The predictor 106 has the same transfer function as the long term predictor 203 of Figure 2.

Step 307: In this step, the initial state of the perceptual filter 107 is set to the value FS supplied by the initial state extractor 110. The difference $R-E$ calculated by a subtractor 121 (Figure 1) is then supplied to the perceptual filter 107 to obtain at the output of the latter filter a target block signal X . As illustrated in Figure 1, the STP parameters are applied to the filter 107 to vary its transfer function in relation to these parameters. Basically, $X = S' - P$ where P represents the contribution of the long term prediction (LTP) including "ringing" from the past excitations. The MSE criterion which applies to Δ can now be stated in the following matrix notations.

$$\begin{aligned} \min_k |\Delta|^2 &= \min_k |S' - \hat{S}|^2 = \min_k |S' - [P - gA_k H^T]|^2 \\ &= \min_k |X - gA_k H^T|^2 \end{aligned} \quad (6)$$

where H accounts for the global filter transfer function $F(z)/(1-B(z))A(z)^{-1}$. It is an $L \times L$ lower triangular Toeplitz matrix formed from the $h(n)$ response.

5

Step 308: This is the backward filtering step performed by the filter 108 of Figure 1. Setting to zero the derivative of the above equation (6) with respect to the code gain g yields to the optimum gain as follows:

10

$$\frac{\partial \|\Delta\|^2}{\partial g} = 0$$

$$g = \frac{X(A_k H^T)^T}{\|A_k H^T\|^2} \quad (7)$$

15

With this value for g the minimization becomes:

20

$$\min_k \|\Delta\|^2 = \min_k \left\{ \|X\|^2 - \frac{(X(A_k H^T)^T)^2}{\|A_k H^T\|^2} \right\}$$

$$= \max_k \frac{(X(A_k H^T)^T)^2}{\|A_k H^T\|^2} \quad (8)$$

$$= \max_k \frac{((XH)A_k^T)^2}{\alpha_k^2} = \max_k \frac{(DA_k^T)^2}{\alpha_k^2}$$

where $D = (XH)$ and $\alpha_k^2 = \|A_k H^T\|^2$.

30

In step 308, the backward filtered target signal $D=(XH)$ is computed. The term "backward filtering" for this operation comes from the interpretation of (XH) as the filtering of time-reversed X .

Step 309: In this step performed by the optimizing controller 109 of Figure 1, equation (8) is optimized by computing the ratio $(DA_k^T/\alpha_k)^2 = P^2_k/\alpha^2_k$ for each sparse algebraic codeword A_k . The denominator is given by the expression:

$$\alpha^2_k = \|A_k H^T\|^2 = A_k H^T H A_k^T = A_k U A_k^T \quad (9)$$

where U is the Toeplitz matrix of the autocorrelations defined in equation (5c). Calling $S(i)$ and $p(i)$ respectively the amplitude and position of the i th non zero impulse ($i = 1, 2, \dots, N$), the numerator and (squared) denominator simplify to the following:

$$DA_k^T = \sum_{i=1}^N S(i)D(p_i) \quad (10a)$$

$$\alpha^2_k = \sum_{i=1}^N S^2(i)U(p_i, p_i) + 2 \sum_{i=1}^{N-1} \sum_{j=i+1}^N S(i)S(j)U(p_i, p_j) \quad (10b)$$

30

where $P(N) = DA_k^T$

A very fast procedure for calculating the above defined ratio for each codeword A_k is described in Figure 5 as a set of N embedded computation loops, N being the number of non zero impulses in the codewords. The quantities $S^2(i)$ and $SS(i,j) = S(i)S(j)$, for $i=1, 2, \dots, N$ and $i < j \leq N$ are pre-stored for maximum speed. Prior to the computations, the values for P_{opt}^2 and α_{opt}^2 are initialized to zero and some large number, respectively. As can be seen in Figure 5, partial sums of the numerator and denominator are calculated in each one of the outer and inner loops, while in the inner loop the largest ratio $P^2(N)/\alpha^2(N)$ is retained as the ratio P_{opt}^2/α_{opt}^2 . The calculating procedure is believed to be otherwise self-explanatory from Figure 5. When the N embedded loops are completed, the code gain is computed as $g = P_{opt} / \alpha_{opt}^2$ (cf. equation (7)). The gain is then quantized, the index k is computed from stored impulse positions using the expression (4), and the L components of the scaled optimum code gC_k are computed as follows:

$$gC_k(n) = g \sum_{i=1}^N f(n-p_i) \quad ; 1 \leq n \leq L \quad (11)$$

with $f(n) = 0$; for $n < 1$

Step 310: The global signal excitation signal $E + gC_k$ is computed by an adder 120 (Figure 1). The initial state extractor module 110, constituted by a

perceptual filter with a transfer function $1/A(z\gamma^{-1})$ varying in relation to the STP parameters, subtracts from the residual signal R the signal excitation signal $E + gCk$ for the sole purpose of obtaining the
5 final filter state FS for use as initial state in filter 107 and module 104.

Step 311: The set of four parameters STP, LTP, k and g are converted into the proper digital channel format
10 by a multiplexer 111 completing the procedure for encoding a block S of samples of speech signal.

Accordingly, the present invention provides a fully quantized Algebraic Code Excited Linear
15 Prediction (ACELP) vocoder giving near toll quality at rates ranging from 4 to 16 kbits. This is achieved through the use of the above described dynamic codebook and associated fast search algorithm.

20 The drastic complexity reduction that the present invention offers when compared to the prior art techniques comes from the fact that the search procedure can be brought back to Ak -code space by a modification of the so called backward filtering
25 formulation. In this approach the search reduces to finding the index k for which the ratio $|DAk^T|/\alpha_k$ is the largest. In this ratio, Ak is a fixed target signal and α_k is an energy term the computation of which can be done with very few operations by codeword
30 when N , the number of non zero components of the codeword Ak , is small.

Although a preferred embodiment of the present invention has been described in detail hereinabove, this embodiment can be modified at will, within the scope of the appended claims, without departing from the nature and spirit of the invention. As an example, many types of algebraic codes can be chosen to achieve the same goal of reducing the search complexity while many types of coloring filters can be used. Also the invention is not limited to the treatment of a speech signal; other types of sound signal can be processed. Such modifications, which retain the basic principle of combining an algebraic code generator with a coloring filter, are obviously within the scope of the subject invention.

The embodiments of the invention in which an exclusive property or privilege is claimed are defined as follows.

1. A method of producing an excitation signal that can be used in synthesizing a sound signal, comprising the steps of:

generating a codeword signal in response to an index signal associated to said codeword signal, said signal generating step using an algebraic code to generate the said codeword signal; and

filtering the so generated codeword signal to produce said excitation signal.

2. A method as defined in claim 1, in which the algebraic code is a sparse algebraic code.

3. A method as defined in claim 1, wherein the excitation signal has frequency characteristics, and wherein said filtering step comprises processing the codeword signal through a coloring filter having a transfer function varying in time to thereby shape the frequency characteristics of the excitation signal so as to damp frequencies perceptually annoying the human ear.

4. A method as defined in claim 3, in which the transfer function of the coloring filter is varied in relation to linear predictive coding parameters representative of spectral characteristics of the said sound signal.

5. A dynamic codebook for producing an excitation signal that can be used in synthesizing a sound signal, comprising:

means for generating a codeword signal in response to an index signal associated to said codeword signal, said signal generating means using an algebraic code to generate the said codeword signal; and

means for filtering the so generated codeword signal to produce said excitation signal.

6. A codebook as defined in claim 5, in which the algebraic code is a sparse algebraic code.

7. A codebook as defined in claim 5, wherein the excitation signal has frequency characteristics, and wherein said filtering means comprises a coloring filter having a transfer function varying in time to shape the frequency characteristics of the excitation signal so as to damp frequencies perceptually annoying the human ear.

8. A codebook as defined in claim 7, in which the coloring filter comprises an input supplied with linear predictive coding parameters representative of spectral characteristics of the said sound signal to vary the said transfer function.

9. A method of selecting one particular algebraic codeword that can be processed to produce a signal excitation for a synthesis means capable of synthesizing a sound signal, comprising the steps of:

whitening said sound signal to be synthesized to generate a residual signal;

computing a target signal X by processing a difference between the said residual signal and a long term prediction component of said signal excitation;

backward filtering the target signal to calculate a value D of the said target signal in the domain of an algebraic code;

calculating, for each codeword among a plurality of available algebraic codewords A_k expressed in the said algebraic code, a target ratio which is function of the value D , the codeword A_k , and a transfer function $H = D / X$; and

selecting the said one particular codeword among said plurality of available algebraic codewords in function of the calculated target ratios.

10. The selecting method of claim 9, in which said target ratio comprises a numerator given by

the expression $P^2(k) = (DAk^T)^2$ and a denominator given by the expression $\alpha^2 k = \|AkH\|^2$, where Ak and H are under the form of matrix.

11. The selecting method of claim 10, wherein each codeword Ak is a waveform comprising a small number of non-zero impulses each of which can occupy different positions in the waveform to thereby enable composition of different codewords.

12. The selecting method of claim 11, in which said target ratio calculating step uses a calculating procedure including embedded loops in which are calculated contributions of the non-zero impulses of the considered algebraic codeword to the said numerator and denominator and in which the so calculated contributions are added to previously calculated sum values of said numerator and denominator, respectively.

13. The selecting method of claim 12, wherein the embedded loops comprise an inner loop, and wherein the said codeword selecting step comprises the steps of:

processing in the inner loop the said calculated target ratios to determine an optimized target ratio; and

selecting the said one particular algebraic codeword in function of said optimized target ratio.

14. The selecting method of claim 9, wherein the said codeword selecting step comprises the steps of:

processing the said calculated target ratios to determine an optimized target ratio; and

selecting the said one particular algebraic codeword in function of said optimized target ratio.

15. An encoder for selecting one particular algebraic codeword that can be processed to produce a signal excitation for a synthesis means capable of synthesizing a sound signal, comprising:

means for whitening said sound signal to be synthesized and thereby generating a residual signal;

means for computing a target signal X by processing a difference between the said residual signal and a long term prediction component of said signal excitation;

means for backward filtering the target signal to calculate a value D of the said target signal in the domain of an algebraic code;

means for calculating, for each codeword among a plurality of available algebraic codewords A_k expressed in the said algebraic code, a

target ratio which is function of the value D , the codeword A_k , and a transfer function $H = D / X$; and means for selecting the said one particular codeword among said plurality of available algebraic codewords in function of the calculated target ratios.

16. The encoder of claim 15, in which said target ratio comprises a numerator given by the expression $P^2(k) = (DAk^T)^2$ and a denominator given by the expression $\alpha^2 k = |AkH|^2$, where A_k and H are under the form of matrix.

17. The encoder of claim 16, wherein each codeword A_k is a waveform comprising a small number of non-zero impulses each of which can occupy different positions in the waveform to thereby enable composition of different codewords.

18. The encoder of claim 17, in which said target ratio calculating means comprises means for calculating into a plurality of embedded loops contributions of the non-zero impulses of the considered algebraic codeword to the said numerator and denominator and for adding the so calculated contributions to previously calculated sum values of said numerator and denominator, respectively.

19. The encoder of claim 18, wherein the embedded loops comprise an inner loop, and wherein the said codeword selecting means comprises:

means for processing in the inner loop the said calculated target ratios to determine an optimized target ratio; and

means for selecting the said one particular algebraic codeword in function of said optimized target ratio.

20. The encoder of claim 15, wherein the said codeword selecting means comprises:

means for processing the said calculated target ratios to determine an optimized target ratio; and

means for selecting the said one particular algebraic codeword in function of said optimized target ratio.

21. A method of generating at least one long term prediction parameter related to a sound signal in view of encoding the said sound signal, comprising the steps of:

whitening said sound signal to generate a residual signal;

producing a long term prediction component of a signal excitation for a synthesis means capable of synthesizing the said sound signal, said producing step including estimating an unknown portion

of the long term prediction component with the said residual signal; and

calculating the said at least on long term prediction parameter in function of the so produced long term prediction component of said signal excitation.

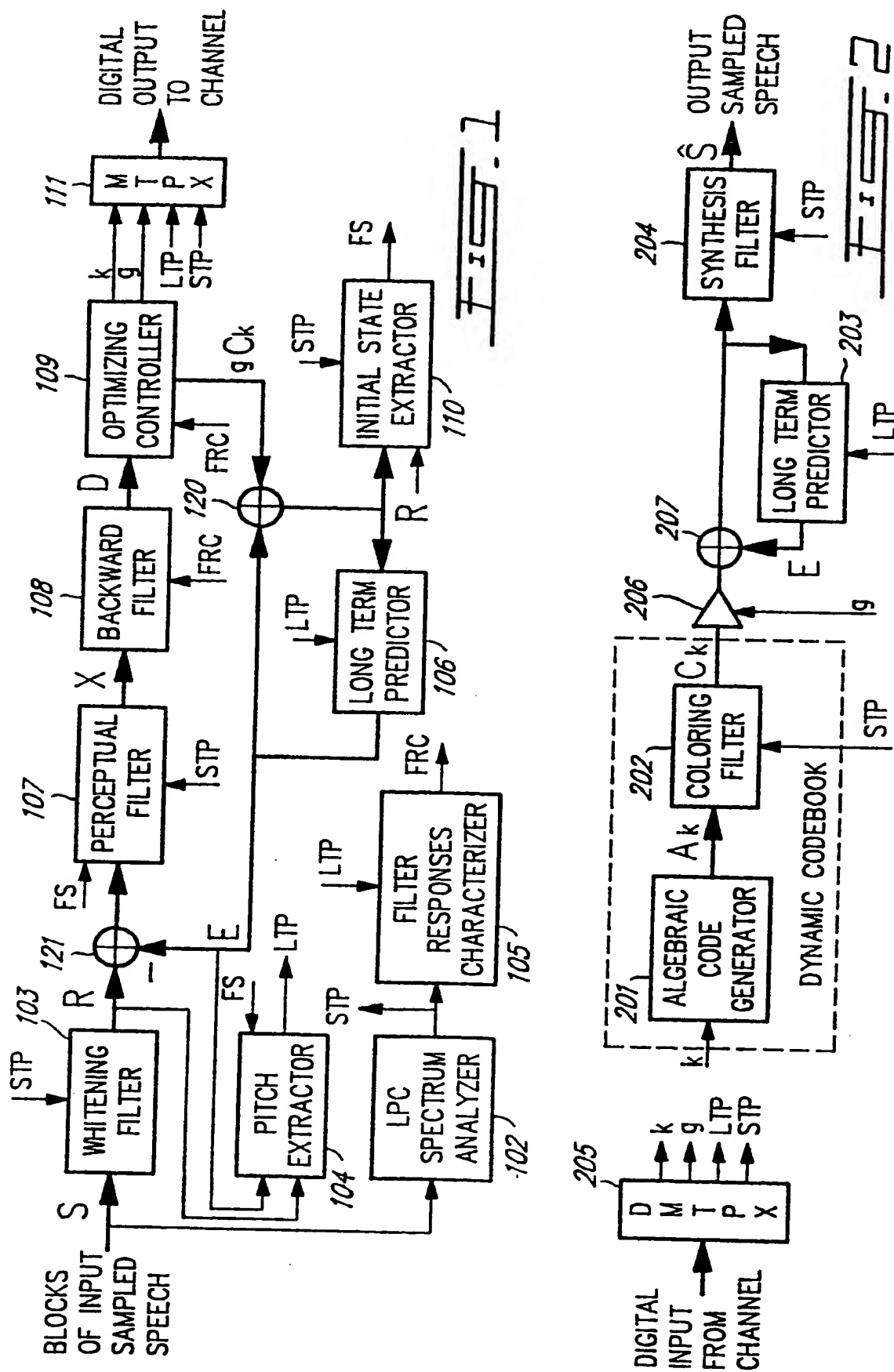
22. A device for generating at least one long term prediction parameter related to a sound signal in view of encoding the said sound signal, comprising:

means for whitening said sound signal and thereby generating a residual signal;

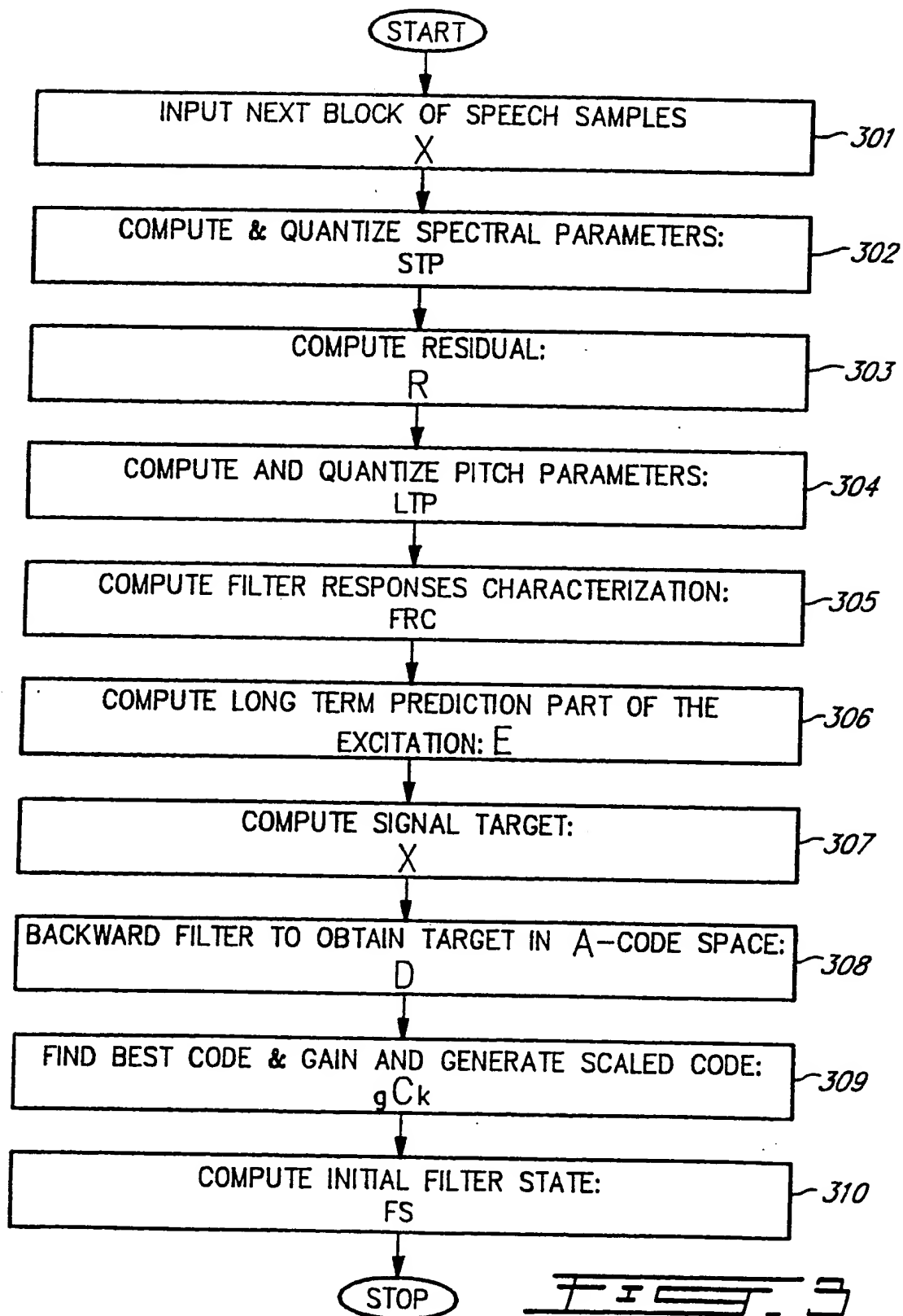
means for producing a long term prediction component of a signal excitation for a synthesis means capable of synthesizing the said sound signal, said producing means including means for estimating an unknown portion of the long term prediction component with the said residual signal; and

means for calculating the said at least one long term prediction parameter in function of the so produced long term prediction component of said signal excitation.

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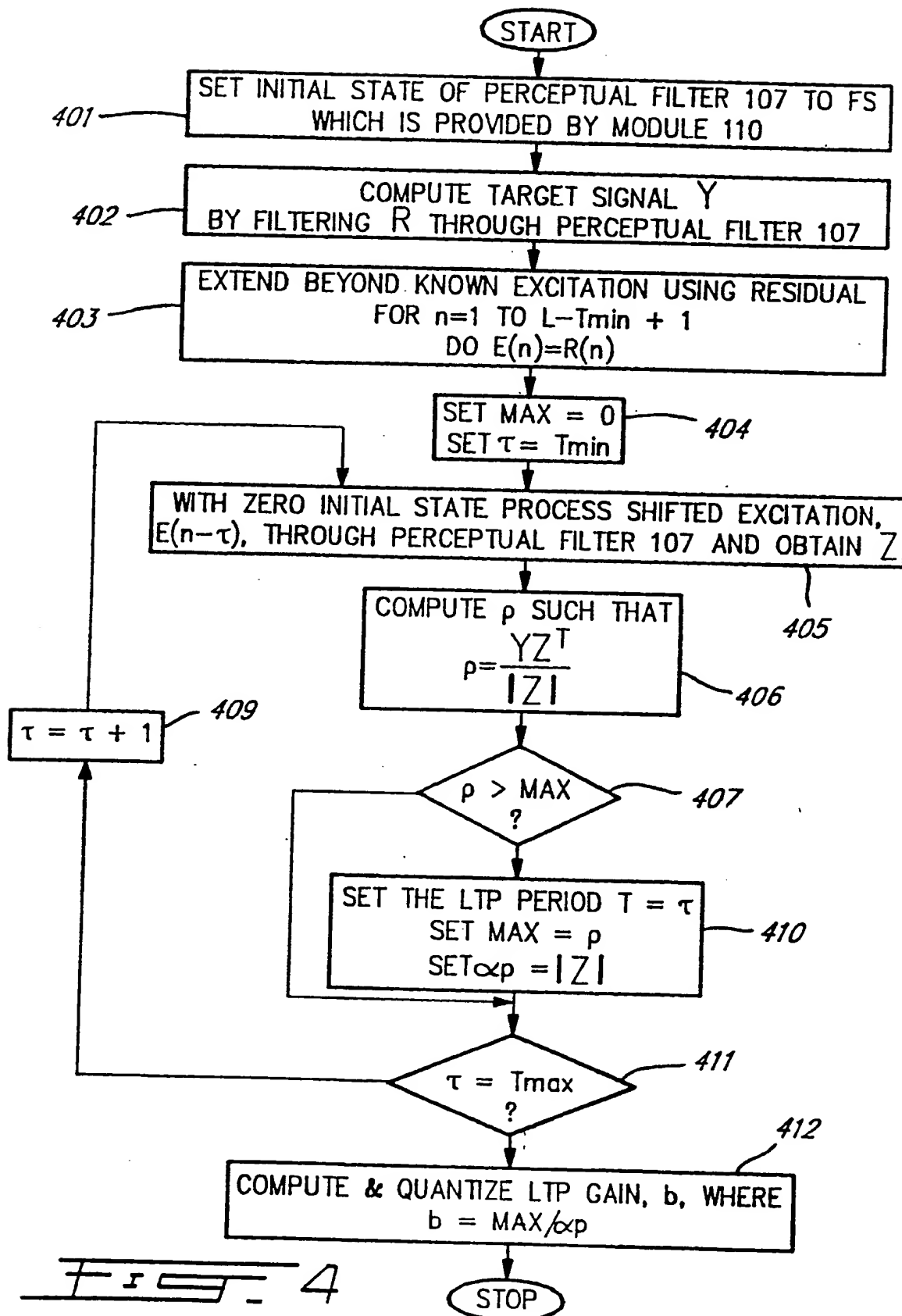


Fig. 5

FOR EACH POSITION, p1, OF IMPULSE S(1)....DO

- $P(1)=S(1)D(p_1)$
- $\alpha^2(1)=S^2(1)U(p_1, p_1)$

• FOR EACH POSITION, p2, OF IMPULSE S(2)....DO

- $P(2)=P(1)+S(2).D(p_2)$
- $\alpha^2(2)=\alpha^2(1)+S^2(2)U(p_2, p_2)+2SS(1,2)U(p_1, p_2)$

• FOR EACH POSITION...

• FOR EACH POSITION, pn, OF IMPULSE S(N)....DO

- $P(N)=P(N-1)+S(N)D(p_N)$
- $\alpha^2(N)=\alpha^2(N-1)+S^2(N)U(p_N, p_N)+2\sum_{j=1}^{N-1}SS(j, N)U(p_j, p_N)$
- IF $P^2(N)\alpha^2_{opt} > P^2_{opt}\alpha^2(N)$

THEN

- $P^2_{opt}=P^2(N)$
- $P_{opt}=P(N)$
- $\alpha^2_{opt}=\alpha^2(N)$
- MEMORIZE N PULSE POSITIONS

INTERNATIONAL SEARCH REPORT

International Application No PCT/CA 90/00381

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) ⁶ According to International Patent Classification (IPC) or to both National Classification and IPC IPC ⁵ : G 10 L 9/14											
II. FIELDS SEARCHED <div style="text-align: center; margin-top: 10px;">Minimum Documentation Searched ⁷</div> <div style="display: flex; justify-content: space-between;"> Classification System: ⁸ Classification Symbols </div> <div style="margin-top: 10px;"> IPC⁵ G 10 L 9/14 </div> <div style="text-align: center; margin-top: 10px; font-size: small;"> Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched ⁹ </div>											
III. DOCUMENTS CONSIDERED TO BE RELEVANT ¹⁰ <table border="1" style="width: 100%; border-collapse: collapse; margin-top: 5px;"> <thead> <tr> <th style="width: 10%;">Category ¹¹</th> <th style="width: 70%;">Citation of Document, ¹¹ with indication, where appropriate, of the relevant passages ¹²</th> <th style="width: 20%;">Relevant to Claim No. ¹³</th> </tr> </thead> <tbody> <tr> <td style="text-align: center; vertical-align: top;">A</td> <td style="vertical-align: top;"> ICASSP 86, IEEE-IECEJ-ASJ International Conference on Acoustics, Speech, and Signal Processing, 7-11 April 1986, Tokyo, Japan, IEEE (New York, US), A. le Guyader et al.: "A robust 16 kbits/s vector adaptive predictive coder for mobile communications", pages 857-860 see figure 1 <div style="text-align: center;">--</div> </td> <td style="vertical-align: top; text-align: center;">1,3-5,7,8</td> </tr> <tr> <td style="text-align: center; vertical-align: top;">A</td> <td style="vertical-align: top;"> IEEE Global Telecommunications Conference & Exhibition, 28 November - 1 December 1988, Hollywood, Florida, US, volume 1, IEEE, (New York, US), F.F. Tzeng: "Multipulse excitation codebook design and fast search methods for CELP speech coding", pages 590-594 <div style="text-align: center;">--</div> <div style="text-align: right;">./.</div> </td> <td style="vertical-align: top; text-align: center;">1,5</td> </tr> </tbody> </table>			Category ¹¹	Citation of Document, ¹¹ with indication, where appropriate, of the relevant passages ¹²	Relevant to Claim No. ¹³	A	ICASSP 86, IEEE-IECEJ-ASJ International Conference on Acoustics, Speech, and Signal Processing, 7-11 April 1986, Tokyo, Japan, IEEE (New York, US), A. le Guyader et al.: "A robust 16 kbits/s vector adaptive predictive coder for mobile communications", pages 857-860 see figure 1 <div style="text-align: center;">--</div>	1,3-5,7,8	A	IEEE Global Telecommunications Conference & Exhibition, 28 November - 1 December 1988, Hollywood, Florida, US, volume 1, IEEE, (New York, US), F.F. Tzeng: "Multipulse excitation codebook design and fast search methods for CELP speech coding", pages 590-594 <div style="text-align: center;">--</div> <div style="text-align: right;">./.</div>	1,5
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<div style="display: flex; justify-content: space-between; font-size: x-small;"> <div style="width: 45%;"> <p>¹⁴ Special categories of cited documents:</p> <p>"A" document defining the general state of the art which is not considered to be of particular relevance</p> <p>"E" earlier document but published on or after the international filing date</p> <p>"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)</p> <p>"O" document referring to an oral disclosure, use, exhibition or other means</p> <p>"P" document published prior to the international filing date but later than the priority date claimed</p> </div> <div style="width: 45%;"> <p>"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention</p> <p>"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step</p> <p>"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.</p> <p>"&" document member of the same patent family</p> </div> </div>											
IV. CERTIFICATION <table border="1" style="width: 100%; border-collapse: collapse; margin-top: 5px;"> <tr> <td style="width: 50%; padding: 5px;"> Date of the Actual Completion of the International Search <div style="text-align: center; font-weight: bold;">14th February 1991</div> </td> <td style="width: 50%; padding: 5px;"> Date of Mailing of this International Search Report <div style="text-align: center; font-weight: bold;">13. 03. 91</div> </td> </tr> <tr> <td style="width: 50%; padding: 5px;"> International Searching Authority <div style="text-align: center; font-weight: bold;">EUROPEAN PATENT OFFICE</div> </td> <td style="width: 50%; padding: 5px;"> Signature of Authorized Officer <div style="text-align: center;"> Nuria TORIBIO </div> </td> </tr> </table>			Date of the Actual Completion of the International Search <div style="text-align: center; font-weight: bold;">14th February 1991</div>	Date of Mailing of this International Search Report <div style="text-align: center; font-weight: bold;">13. 03. 91</div>	International Searching Authority <div style="text-align: center; font-weight: bold;">EUROPEAN PATENT OFFICE</div>	Signature of Authorized Officer <div style="text-align: center;"> Nuria TORIBIO </div>					
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III. DOCUMENTS CONSIDERED TO BE RELEVANT (CONTINUED FROM THE SECOND SHEET)		
Category *	Citation of Document, " with indication, where appropriate, of the relevant passages	Relevant to Claim No.
A	ICASSP 87, 1987 International Conference on Acoustics, Speech, and Signal Processing, 6-9 April 1987, Dallas, Texas, US, volume 4, IEEE, (New York, US), J.-P. Adoul et al.: "A comparison of some algebraic structures for CELP coding of speech", pages 1953-1956 -----	2,6

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